IN THE CLAIMS:

Please amend claim 51 as indicated in the following.

Claims Listing:

1. (Previously Presented) A method comprising:
determining a synchronization state of an audio data relative to a system clock;
when the synchronization state is in a first state maintaining a current playback;
when the synchronization state is in a second state making a first playback adjustment to the audio data, wherein the first playback adjustment includes performing a sample rate conversion of one or more audio data samples of the audio data; and
when the synchronization state is in a third state making a second playback adjustment to the audio data, wherein the second playback adjustment provides a coarser playback adjustment than the first playback adjustment.

2. (Previously Presented) The method as in Claim 1, further comprising:
when the synchronization state is in a fourth state initializing the system clock to a predefined value.

3. (Original) The method as in Claim 2, wherein the predefined value is equal to a representation of a system clock associated with a source of data.

4. (Original) The method as in Claim 3, wherein the representation of the system clock associated with the source of the audio data is a program counter clock associated with MPEG-type data.

5.–6. (Canceled).

7. (Previously Presented) The method as in Claim 1, wherein the second playback adjustment includes adjustments to PES packets.

8. (Previously presented) The method as in Claim 7, wherein the second playback adjustment includes repeating PES packets.

9. (Previously presented) The method as in Claim 7, wherein the second playback adjustment includes dropping PES packets.

10. (Original) The method as in Claim 1, wherein determining a synchronization state includes comparing a PTS value to an STC value.

11. (Original) The method as in Claim 10, wherein a difference between the PTS value and the STC value is compared to a delta value.

12. (Original) The method as in Claim11, wherein a delta value associated with the first state indicates a time difference equivalent to playback of a single audio data sample.

13. (Original) The method as in Claim 12, wherein the time to play a single audio sample is calculated by determining an audio data type.

14. (Original) The method as in Claim 13, wherein determining the audio data type is based upon a data stream type.

15. (Original) The method as in Claim 13, wherein determining the audio data type is based upon a data stream identifier.

16. (Original) The method as in Claim 11, wherein a delta value associated with the second state indicates a range of time from the time required for playback of 2 audio samples to the time required for playback of 32 audio data samples.

17. (Previously presented) The method as in Claim 16, wherein the time to play an audio sample is calculated by determining an audio data type.

18. (Original) The method as in Claim 17, wherein determining the audio data type is based upon a data stream type.

19. (Original) The method as in Claim 17, wherein determining the audio data type is based upon a data stream identifier.

20. (Original) The method as in Claim 11, wherein a delta value associated with the third state indicates a range of time from the time required for playback of 1 audio frame to the time required for playback of 3 audio data frames.

21. (Original) The method as in Claim 20, wherein the time required for playing audio frames is calculated by determining an audio data type.

22. (Original) The method as in Claim 21, wherein determining the audio data type is based upon a data stream type.

23. (Original) The method as in Claim 21, wherein determining the audio data type is based upon a data stream identifier.

24. (Original) The method as in Claim 11, wherein the delta value is variable..

25. (Previously presented) A system comprising:

a data processor having an I/O buffer;

a first data processing component capable of maintaining playback of audio data samples according to a first state of synchronization;

a second data processing component capable of:

performing a first playback adjustment to maintain a second state of synchronization, wherein the first playback adjustment includes performing a sample rate conversion of one or more of the audio data samples;

a third data processing component capable of:

performing a second playback adjustment to maintain a third state of synchronization, wherein the second playback adjustment provides a coarser playback adjustment than the first playback adjustment; and

a system time clock capable of tracking presentation times associated with the audio data.

26. (Original) The system as in Claim 25, further including:

a fourth data processing component capable of initializing the system time clock to a predefined value to maintain a fourth state of synchronization.

27. (Original) The system as in Claim 26, wherein the predefined value is equal to a representation of a system clock associated with a source of data.

28. (Original) The system as in Claim 27, wherein the representation of the system clock associated with the source of the audio data is a program counter clock associated with MPEG-type data.

29. – 30. (Canceled)

31. (Original) The system as in Claim 25, wherein the second playback adjustment includes adjustments to PES packets.

32. (Original) The system as in Claim 31, wherein the second playback adjustment includes repeating PES packets.

33. (Original) The system as in Claim 31, wherein the second playback adjustment includes dropping PES packets.

34. (Original) The system as in Claim 25, wherein operations within the data processing components are determined by comparing a PTS value to an STC value.

35. (Original) The system as in Claim 34, wherein a difference between the PTS value and the STC value is compared to a delta value.

36. (Original) The system as in 35, wherein a delta value associated with the first state indicates a time difference equivalent to playback of a single audio data sample.

37. (Previously presented) The system as in Claim 36, wherein the time to play a single audio sample is calculated based on a determination of an audio data type.

38. (Original) The system as in Claim 35, wherein a delta value associated with the second state indicates a range of time from the time required for playback of 2 audio samples to the time required for playback of 32 audio data samples.

39. (Previously presented) The system as in Claim 35, wherein a delta value associated with the third state indicates a range of time from the time required for playback of 1 audio frame to the time required for playback of 3 audio frames.

40. (Original) The system as in Claim 35, wherein the delta value is variable.

41. (Previously presented) The system as in Claim 25, further comprising memory operably coupled to the data processor and capable of storing code for the first, second and third data processing components.

42. (Original) The system as in Claim 25, wherein the first, second and third data processing components are represented in hardware.

43. (Previously presented) The system as in Claim 25, wherein the first data processing component includes a demultiplexer.

44. (Original) The system as in Claim 25, wherein the presentation time is determined through a presentation time stamp associated with a data stream.

45. (Original) The system as in Claim 25, wherein the presentation time of the data packets are interpolated from the presentation time stamp associated with the data stream.

46. (Original) The method as in Claim 25, wherein the second data processing component is further capable of processing data packets into data samples.

47. (Previously presented) A computer readable medium tangibly embodying a plurality of programs of instructions, the plurality of programs including:

a first data processing component capable of maintaining playback of audio data samples according to a first state of synchronization, wherein the first playback adjustment includes performing a sample rate conversion of one or more of the audio data samples;

a second data processing component capable of:

performing a first playback adjustment to maintain a second state of synchronization;

a third data processing component capable of:

performing a second playback adjustment to maintain a third state of synchronization, wherein the second playback adjustment provides a coarser playback adjustment than the first playback adjustment.

48. (Original) The computer readable medium as in Claim 47, further including a fourth data processing component capable of initializing the system time clock to a predefined value to maintain a fourth state of synchronization.

49. (Original) The computer readable medium as in Claim 47, wherein the third data processing component is capable of controlling hardware used to process the data packets into data samples.

50. (Original) The computer readable medium as in Claim 47, wherein the second data processing component is capable of controlling hardware for processing the data samples.

51. (Currently Amended) The ~~system~~ computer readable medium as in Claim 47, wherein operations within the data processing components are determined by comparing a PTS value to an STC value.

52. (Previously presented) A method comprising:

receiving an MPEG-type transport stream;

demultiplexing the MPEG-type transport stream to generate transport packets;

synchronizing a system time clock to a program clock reference received through the MPEG-type transport stream;

determining if a PTS value associated with the transport packets is within a predefined value of the system time clock;

when the PTS value is within the predefined value, performing a sample rate conversion of audio samples related to the transport packets; and

when the PTS value is not within the predefined value, adjusting PES packets related to the transport packets.

53. (Original) The method as in Claim 52, wherein the transport packets are processed into the PES packets.

54. (Original) The method as in Claim 53, wherein the PES packets are processed into the audio samples.

55. (Original) The method as in Claim 52, wherein the predetermined value indicates a range of time from the time required for playback of 1 audio frame to the time required for playback of 3 audio data frames.

56. (Original) The method as in Claim 55, wherein the time required for playing audio frames is calculated by determining an audio data type.